Examples for Using Speech Signal Processing Toolkit Ver. 3.9

SPTK working group

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Contents

1 Basics

1.1 Help message

```
impulse -h
```

1.2 Data type conversion between "little endian" and "big endian."

Files: data.short: speech data included in this example (short integer, 16 kHz sampling, little endian)
data.short-b: speech data (short integer, 16 kHz sampling, big endian)

```
swab +s < data.short > data.short-b
```

1.3 Dump a binary data file

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)

```
dmp +s data.short | less
```

1.4 Data type conversion from "short int" to "float"

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)
data.float: speech data (float, 16 kHz sampling)[1][2]

```
x2x +sf < data.short > data.float
```

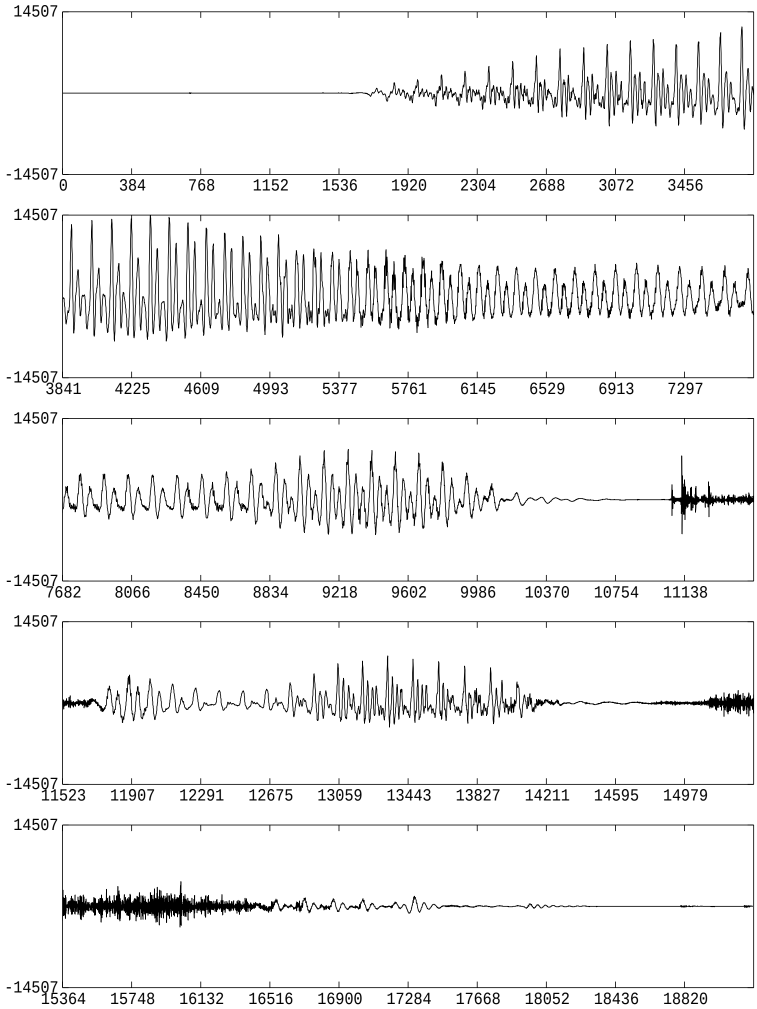
1.5 Plotting speech waveform on X-window

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)

```
gwave +s data.short | xgr
```

[1] By clicking links in this PDF file, your PC may play some speech files, which were converted from "float" format into "wav" format (16 kHz sampling, 16-bit integer).

[2] If you compiled SPTK with "`--enable-double`" option, please use "`+sd`" option instead of "`+sf`" and "`+d`" option instead of "`+f`".



1.6 Save the figure in an Encapsulated PostScript file

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)
figure.eps: Encapsulated PostScript file

```
gwave +s data.short | psgr > figure.eps
```

1.7 Play a sound file

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)

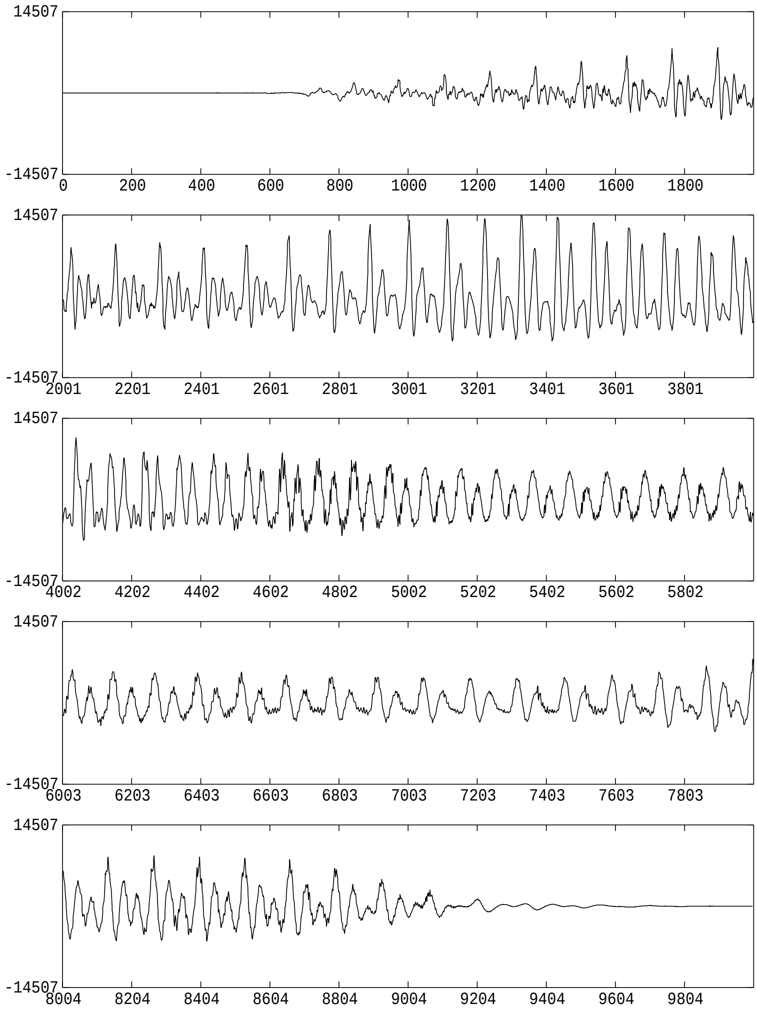
Note: This works only on Linux, Solaris, and FreeBSD.

```
da +s -s 16 -a 100 data.short
```

1.8 Cut a portion out of a file

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)

```
bcut +s -s 1000 -e 11000 < data.short |\
gwave +s | xgr
```



2 Pitch Extraction from Speech Waveform

2.1 A pitch extractor

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)

Conditions: frame period: 80 points (5 ms)
minimum fundamental frequency for search: 80 Hz
maximum fundamental frequency for search: 165 Hz

Note: Options should be adjusted for each speech data.

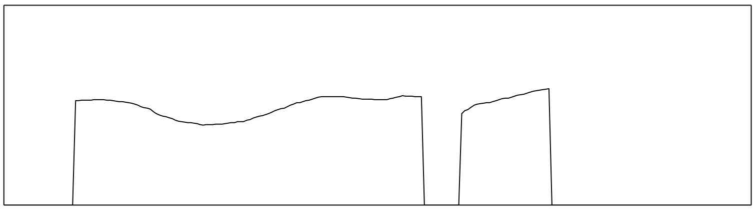
```
x2x +sf data.short | pitch -a 1 -s 16 -p 80 -L 80 -H 165 > data.pitch
```

2.2 Plotting the extracted pitch contour

Files: data.pitch: pitch data extracted from speech data "data.short" (float)

Conditions: Minimum value of vertical axis: 0.0
Maximum value of vertical axis: 250.0
Width: 15 cm
Height: 4 cm

```
fdrw -y 0 250 -W 1.5 -H 0.4 < data.pitch | xgr
```



0 -x- 240 0 -y- 250

3 Speech Analysis/Synthesis Based on Mel-Cepstral Representation

3.1 Mel-cepstral analysis of speech

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)
data.mcep: mel-cepstrum (float)

Conditions: frame length: 400 points (25 ms)
frame period: 80 points (5 ms)
window: Blackman window
analysis order: 20
frequency warping parameter: $\alpha = 0.42$
FFT size: 512 points

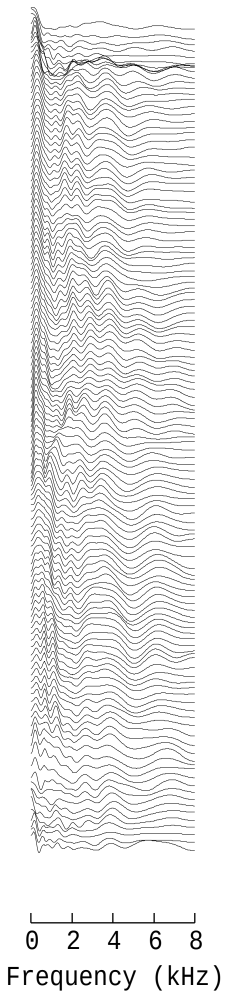
```
x2x +sf < data.short | frame -l 400 -p 80 | window -l 400 -L 512 |\
mcep -l 512 -m 20 -a 0.42 > data.mcep
```

3.2 Plotting spectral estimates from mel-cepstrum

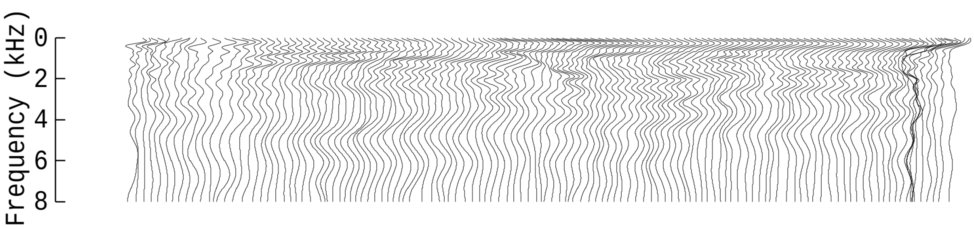
Files: data.mcep: mel-cepstrum (float)

Conditions: analysis order: 20
frequency warping parameter: $\alpha = 0.42$
FFT size: 512 points
plotted frames: from 10-th to 135-th
sampling frequency: 16 kHz

```
bcut +f -n 20 -s 10 -e 135 < data.mcep |\
mgc2sp -m 20 -a 0.42 -g 0 -l 512 | grlogsp -l 512 -x 8 | xgr
```




```
bcut +f -n 20 -s 10 -e 135 < data.mcep |\
mgc2sp -m 20 -a 0.42 -g 0 -l 512 | grlogsp -l 512 -x 8 -t | xgr
```



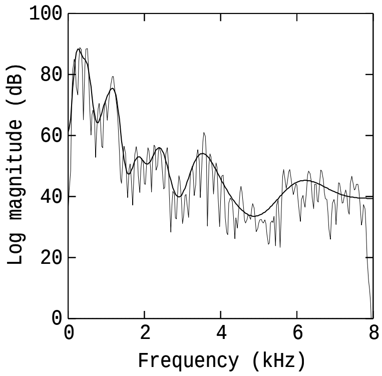
3.3 Plotting the spectral estimate with the FFT spectrum

Files: data.mcep: mel-cepstrum (float)

Conditions: analysis order: 20
frequency warping parameter: $\alpha = 0.42$
FFT size: 512 points

plotted frame: 65-th
sampling frequency: 16 kHz

```
( x2x +sf < data.short | frame -l 400 -p 80 | \
bcut +f -l 400 -s 65 -e 65 |\
window -l 400 -L 512 | spec -l 512 |\
glogsp -l 512 -x 8 -p 2 ;\
\
bcut +f -n 20 -s 65 -e 65 < data.mcep |\
mgc2sp -m 20 -a 0.42 -g 0 -l 512 | glogsp -l 512 -x 8 ) | xgr
```



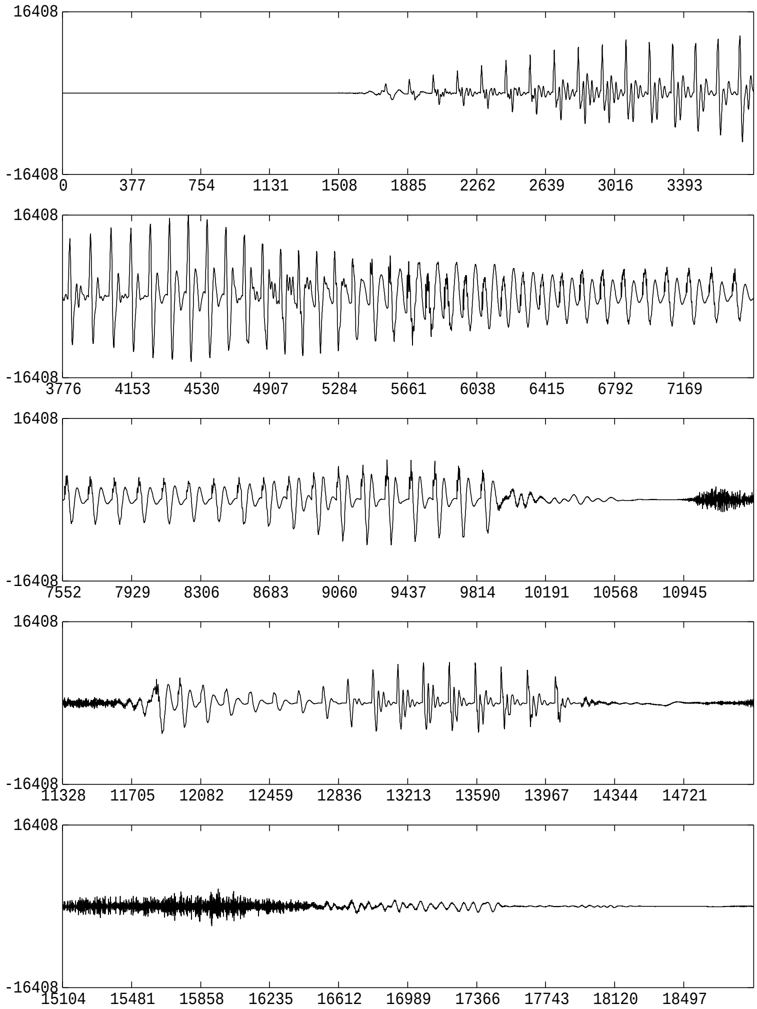
3.4 Speech synthesis from mel-cepstrum

Files: data.pitch: pitch data extracted from speech data ”data.short” (float)
data.mcep: mel-cepstrum (float)
data.mcep.syn: synthesized speech (float)

Conditions: frame period: 80 points (5 ms)
analysis order: 20
frequency warping parameter: $\alpha = 0.42$

```
excite -p 80 data.pitch |\
mlsadf -m 20 -a 0.42 -p 80 data.mcep > data.mcep.syn

gwave data.mcep.syn | xgr
```




```
da +f -s 16 data.mcep.syn
```

4 Speech Analysis/Synthesis based on LPC

4.1 LPC analysis of speech

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)
data.lpc: LPC coefficients (float)

Conditions: frame length: 400 points (25 ms)
frame period: 80 points (5 ms)
window: Blackman window
analysis order: 20

```
x2x +sf < data.short | frame -l 400 -p 80 | window -l 400 |\
lpc -l 400 -m 20 > data.lpc
```

4.2 Plotting spectral estimates from LPC coefficients

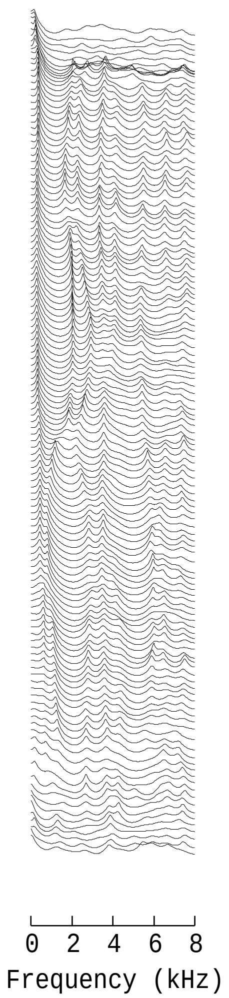
Files: data.lpc: LPC coefficients (float)

Conditions: analysis order: 20

```
bcut +f -n 20 -s 10 -e 135 < data.lpc |\
spec -l 512 -n 20 | grlogsp -l 512 -x 8 | xgr
```

or

```
bcut +f -n 20 -s 10 -e 135 < data.lpc |\
mgc2sp -m 20 -a 0 -g -1 -n -u -l 512 |\
grlogsp -l 512 -x 8 | xgr
```

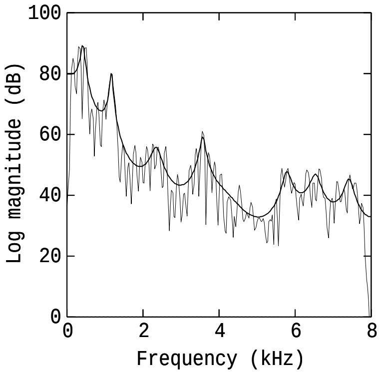


4.3 Plotting the spectral estimate with the FFT spectrum

Files: data.lpc: LPC coefficients (float)

Conditions: analysis order: 20
plotted frame: 65-th
sampling frequency: 16 kHz

```
( x2x +sf < data.short | frame -l 400 -p 80 | \
bcut +f -l 400 -s 65 -e 65 |\
window -l 400 -L 512 | spec -l 512 |\
glogsp -l 512 -x 8 -p 2 ;\
\
bcut +f -n 20 -s 65 -e 65 < data.lpc > data.tmp ;\
spec -l 512 -n 20 -p data.tmp | glogsp -l 512 -x 8 ;\
\rm data.tmp ) | xgr
```



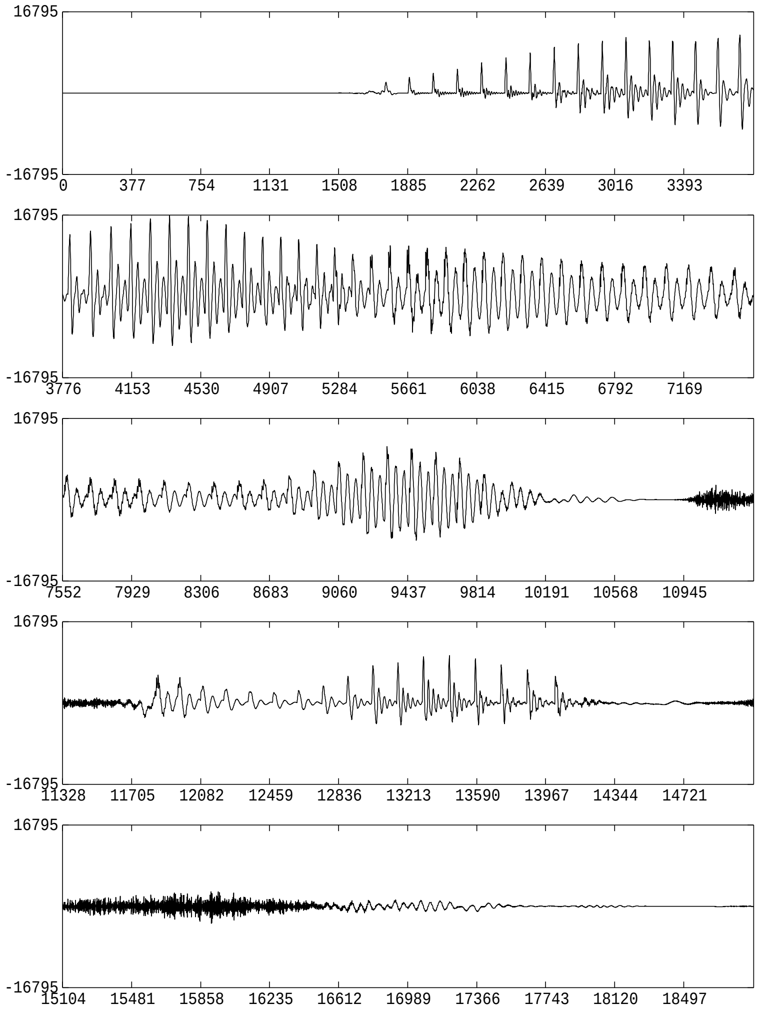
4.4 Speech synthesis from LPC coefficients

Files: data.pitch: pitch data extracted from speech data "data.short" (float)
data.lpc: LPC coefficients (float)
data.lpc.syn: synthesized speech (float)

Conditions: frame period: 80 points (5 ms)
analysis order: 20

```
excite -p 80 data.pitch | poledf -m 20 -p 80 data.lpc > data.lpc.syn
```

```
gwave +f data.lpc.syn | xgr
```




```
da +f -s 16 data.lpc.syn
```

4.5 Obtain PARCOR coefficients from LPC coefficients

Files: data.lpc: LPC coefficients (float)
data.par: PARCOR coefficients (float)

Conditions: analysis order: 20

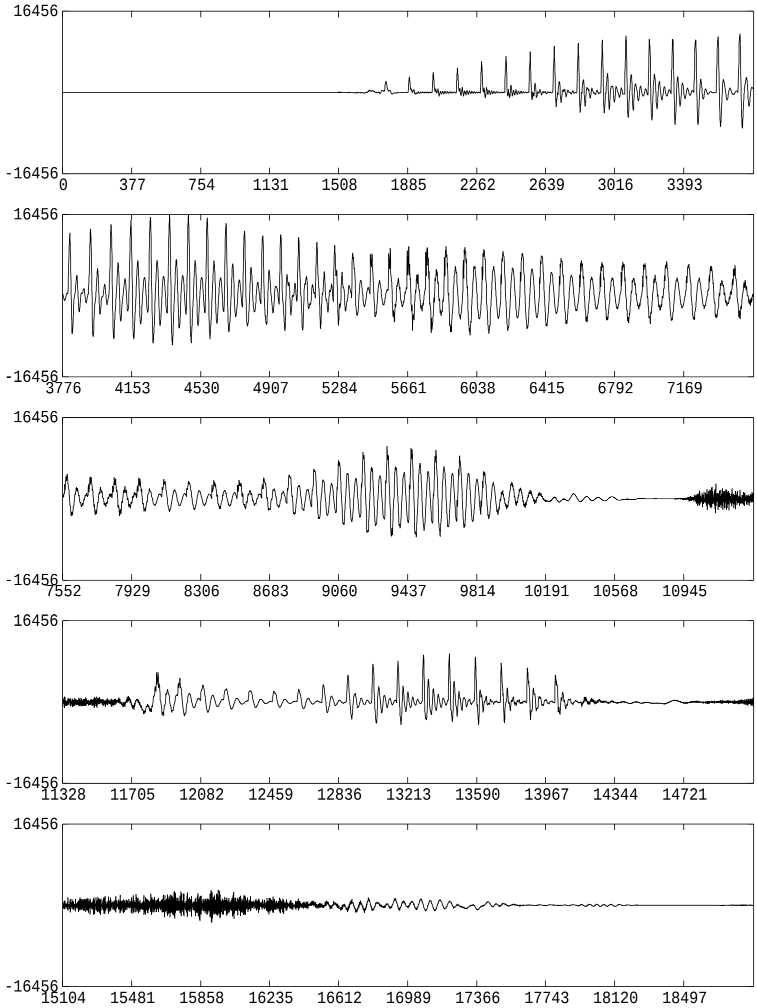
```
lpc2par -m 20 < data.lpc > data.par
```

4.6 Speech synthesis from PARCOR coefficients

Files: data.pitch: pitch data extracted from speech data ”data.short” (float)
data.par: PARCOR coefficients (float)
data.par.syn: synthesized speech (float)

Conditions: frame period: 80 points (5 ms)
analysis order: 20

```
excite -p 80 data.pitch | ltcdf -m 20 -p 80 data.par > data.par.syn
gwave +f data.par.syn | xgr
```



4.7 Obtain LSP coefficients from LPC coefficients

Files: data.lpc: LPC coefficients (float)
data.lsp: LSP coefficients (float)

Conditions: analysis order: 20
split number of unit circle: 256

```
lpc2lsp -m 20 -n 256 < data.lpc > data.lsp
```

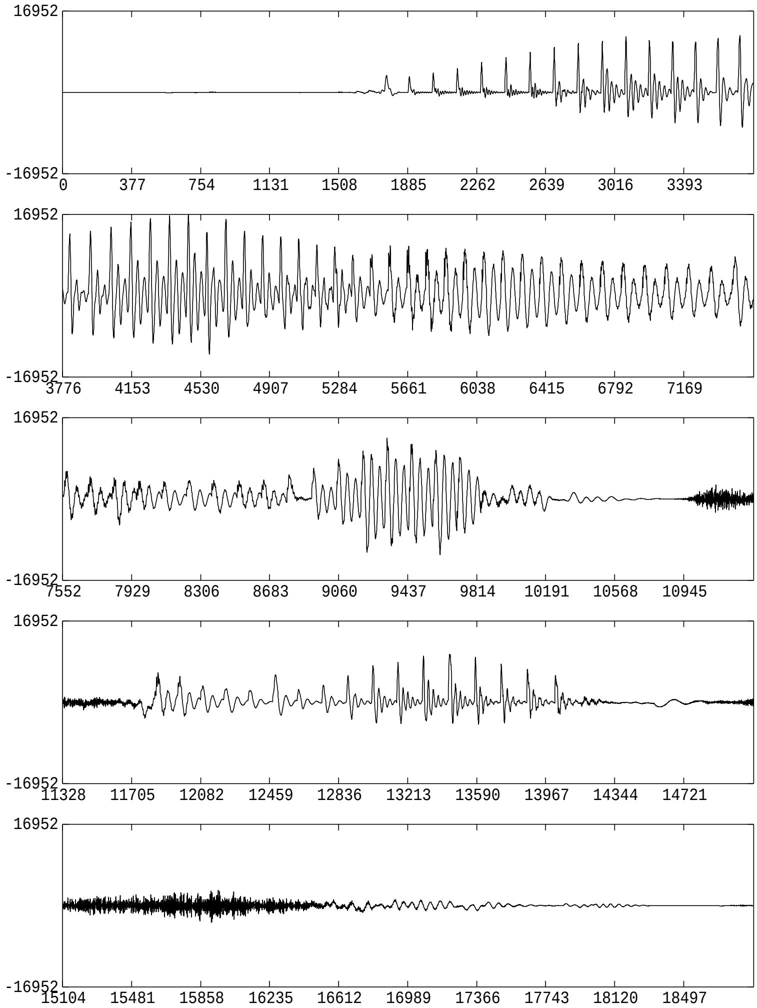
4.8 Speech synthesis from LSP coefficients

Files: data.pitch: pitch data extracted from speech data "data.short" (float)
data.lsp: LSP coefficients (float)
data.lsp.syn: synthesize speech (float)

Conditions: frame period: 80 points (5 ms)
analysis order: 20

```
excite -p 80 data.pitch | lspdf -m 20 -p 80 data.lsp > data.lsp.syn
```

```
gwave +f data.lsp.syn | xgr
```




```
da +f -s 16 data.lsp.syn
```

5 Speech Analysis/Synthesis Based on Mel-Generalized Cepstral Representation

5.1 Mel-generalized cepstral analysis of speech

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)
data.mgcep: mel-generalized cepstrum (float)

Conditions: frame length: 400 points (25 ms)
frame period: 80 points (5 ms)
window: Blackman window
analysis order: 20
frequency warping parameter: $\alpha = 0.42$
power parameter: $\gamma = -1/2$

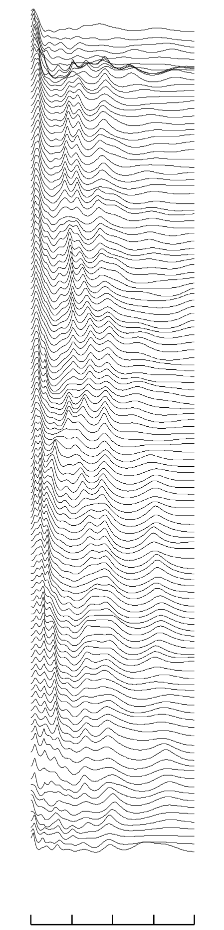
```
x2x +sf < data.short | frame -l 400 -p 80 | window -l 400 -L 512 |\
mgcep -m 20 -a 0.42 -c 2 -l 512 > data.mgcep
```

5.2 Plotting spectral estimates from mel-generalized cepstrum

Files: data.mgcep: mel-generalize cepstrum (float)

Conditions: analysis order: 20
frequency warping parameter: $\alpha = 0.42$
power parameter: $\gamma = -1/2$
plotted frames: from 10-th to 135-th
sampling frequency: 16 kHz

```
bcut +f -n 20 -s 10 -e 135 < data.mgcep |\
mgc2sp -m 20 -a 0.42 -c 2 -l 512 | grlogsp -l 512 -x 8 | xgr
```



5.3 Plotting the spectral estimate with the FFT spectrum

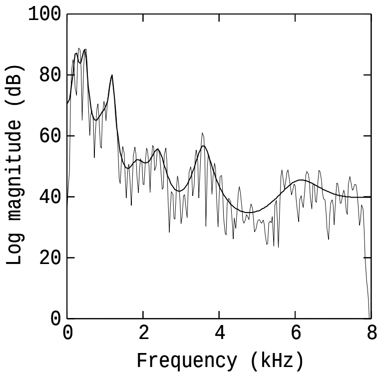
Files: data.mgcep: mel-generalized cepstrum (float)

Conditions: analysis order: 20
frequency warping parameter: $\alpha = 0.42$
power parameter: $\gamma = -1/2$
FFT size: 512 points
plotted frame: 65-th
sampling frequency: 16 kHz

vvvv

```
( x2x +sf < data.short | frame -l 400 -p 80 | \
bcut +f -l 400 -s 65 -e 65 |\
window -l 400 -L 512 | spec -l 512 |\
glogsp -l 512 -x 8 -p 2 ;\
\
bcut +f -n 20 -s 65 -e 65 < data.mgcep |\
```

```
mgc2sp -m 20 -a 0.42 -c 2 -l 512 | glogsp -l 512 -x 8 ) | xgr
```



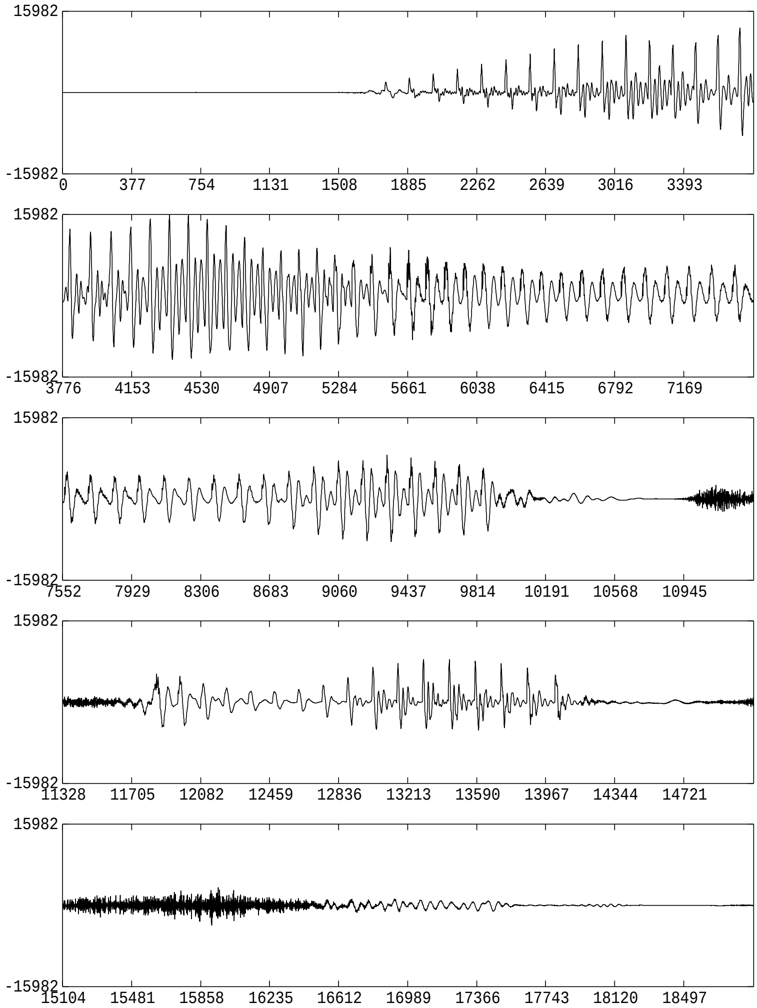
5.4 Speech synthesis from mel-generalized cepstrum

Files: data.pitch: pitch data extracted from speech data ”data.short” (float)
data.mgcep: mel-generalized cepstrum (float)
data.mgcep.syn: synthesized speech (float)

Conditions: frame period: 80 points (5 ms)
analysis order: 20
frequency warping parameter: $\alpha = 0.42$
power parameter: $\gamma = -1/2$

```
excite -p 80 data.pitch |\
mglsadf -m 20 -a 0.42 -c 2 -p 80 data.mgcep > data.mgcep.syn

gwave +f data.mgcep.syn | xgr
```




```
da +f -s 16 data.mgcep.syn
```

6 Vector Quantization of Mel-Cepstrum

6.1 Train a (very small) Codebook

Files: data.mcep: mel-cepstrum for training (float)
codebook.mcep: codebook (float)

Conditions: vector size: 21 (analysis order: 20)
codebook size: 32

```
lbg -n 20 -e 32 < data.mcep > codebook.mcep
```

6.2 Encode (training vectors)

Files: codebook.mcep: codebook (float)
data.mcep.index: index (int)

Conditions: vector size: 21 (analysis order: 20)
codebook size: 32

```
vq -n 20 codebook.mcep < data.mcep > data.mcep.index
```

6.3 Decode (training vectors)

Files: codebook.mcep: codebook (float)
data.mcep.index: index (int)
data.mcep.vq: quantized mel-cepstrum (float)

Conditions: vector size: 21 (analysis order: 20)
codebook size: 32

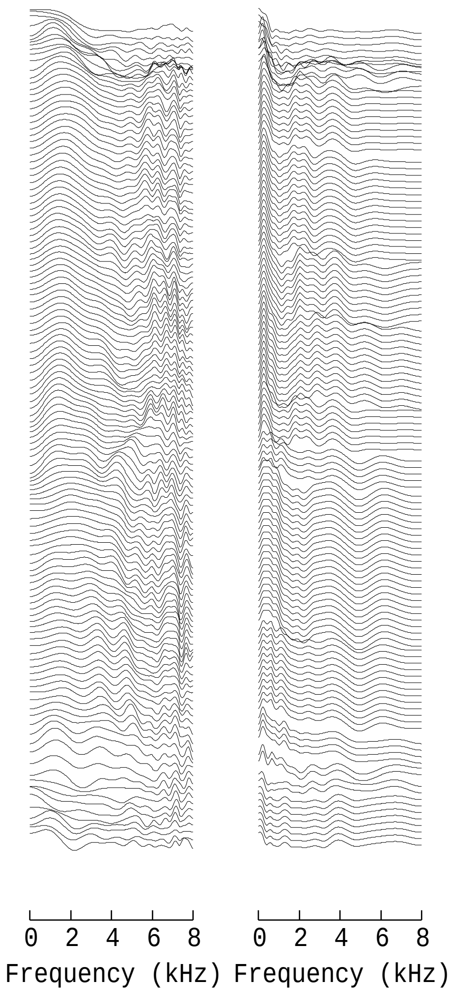
```
ivq -n 20 codebook.mcep < data.mcep.index > data.mcep.vq
```

6.4 Plotting original and quantized spectra

Files: data.mcep: original mel-cepstrum (float)
data.mcep.vq: quantized mel-cepstrum (float)

Conditions: analysis order: 20
frequency warping parameter: $\alpha = 0.42$
plotted frames: from 10-th to 135-th
sampling frequency: 16 kHz

```
( bcut +f -n 20 -s 10 -e 135 < data.mcep |\
mgc2sp -m 20 -a 0.42 -g 0 -l 512 |\
grlogsp -l 512 -x 8 -O 1 -c "(a) original" ;\
\
bcut +f -n 20 -s 10 -e 135 < data.mcep.vq |\
mgc2sp -m 20 -a 0.42 -g 0 -l 512 |\
grlogsp -l 512 -x 8 -O 2 -c "(b) quantized" ) | xgr
```



(a) original (b) quantized

6.5 Performance evaluation on the training data

Files: codebook.mcep: codebook (float)
data.mcep.index: index (int)
data.mcep.vq: quantized vectors (float)
data.mcep.vq.cdist: cepstrum distortion in dB (float)

Conditions: vector size: 21 (analysis order: 20)
codebook size: 32

```
freqt -a 0.42 -m 20 -A 0 -M 255 < data.mcep > data.mcep.cep
freqt -a 0.42 -m 20 -A 0 -M 255 < data.mcep.vq |\
cdist data.mcep.cep -m 255 > data.mcep.vq.cdist
\rm data.mcep.cep
```

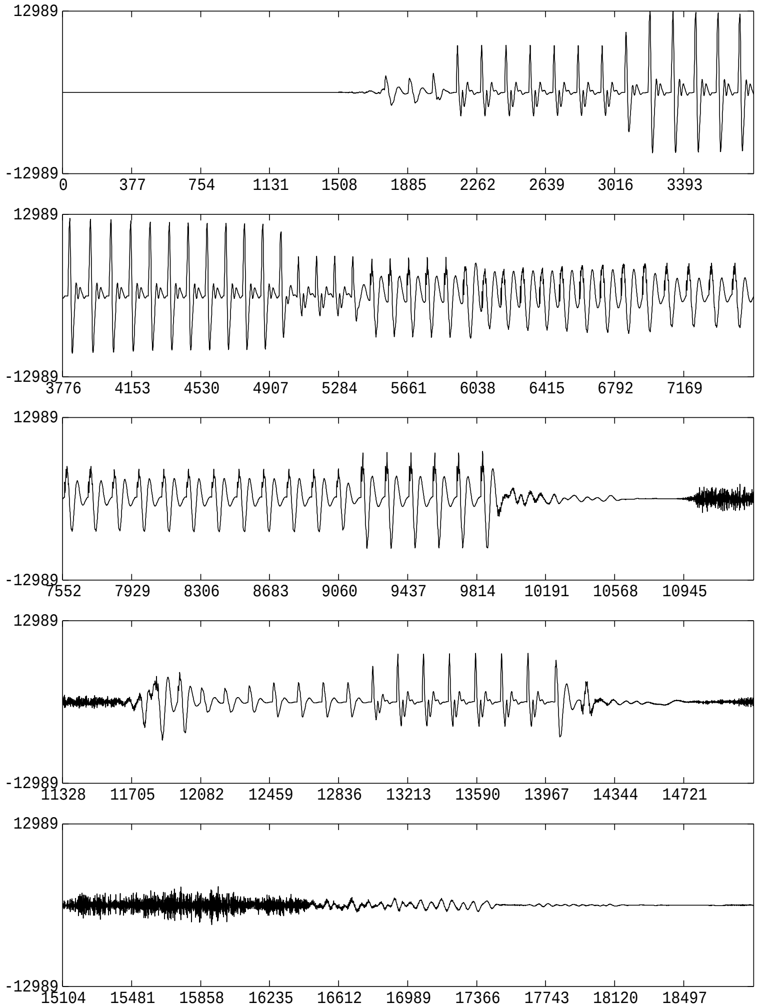
6.6 Speech synthesis from quantized mel-cepstrum

Files: data.pitch: pitch data extracted from speech data "data.short" (float)
data.mcep.vq: quantized mel-cepstrum (float)
data.mcep.vq.syn: synthesized speech (float)

Conditions: frame period: 80 points (5 ms)
analysis order: 20
frequency warping parameter: $\alpha = 0.42$

```
excite -p 80 data.pitch |\
mlsadf -m 20 -a 0.42 -p 80 data.mcep.vq > data.mcep.vq.syn
```

```
gwave +f data.mcep.vq.syn | xgr
```




```
da +f -s 16 data.mcep.vq.syn
```

7 Preparation of Speech Parameter for Speech Recognition

7.1 Cepstrum derived from LPC analysis (LPC cepstrum)

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)

Conditions: frame length: 400 points (25 ms)
frame period: 80 points (5 ms)
window: Blackman window
analysis order: 12
order of LPC cepstrum: 12

```
x2x +sf < data.short | frame -l 400 -p 80 | window -l 400 |\
lpc -l 400 -m 12 | lpc2c -m 12 -M 12 > data.lpc.cep
```

7.2 Mel-cepstrum derived from LPC analysis (LPC mel-cepstrum)

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)

Conditions: frame length: 400 points (25 ms)
frame period: 80 points (5 ms)
window: Blackman window
analysis order: 12
order of LPC mel-cepstrum: 12

```
x2x +sf < data.short | frame -l 400 -p 80 | window -l 400 |\
lpc -l 400 -m 12 |\
lpc2c -m 12 -M 256 |\
freqt -m 256 -a 0 -M 12 -A 0.42 > data.lpc.mcep
```

or

```
x2x +sf < data.short | frame -l 400 -p 80 | window -l 400 |\
lpc -l 400 -m 12 |\
mgc2mgc -m 12 -a 0 -g -1 -n -u -M 12 -A 0.42 -G 0 > data.lpc.mcep
```

7.3 Mel-cepstrum obtained by mel-cepstral analysis

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)
data.mcep: mel-cepstrum (float)

Conditions: frame length: 400 points (25 ms)
frame period: 80 points (5 ms)
window: Blackman window
analysis order: 20
frequency warping parameter: $\alpha = 0.42$
FFT size: 512 points

```
x2x +sf < data.short | frame -l 400 -p 80 | window -l 400 -L 512 |\
mcep -l 512 -m 12 -a 0.42 > data.mcep.mcep
```

7.4 Mel-cepstrum derived from mel-generalized cepstral analysis

Files: data.short: speech data included in this example (short integer, 16 kHz sampling)

Conditions: frame length: 400 points (25 ms)
frame period: 80 points (5 ms)
Blackman window
FFT size: 512 points
(α, γ) for analysis: (0.42, -0.5)
analysis order: 12
order of mel-cepstrum: 12

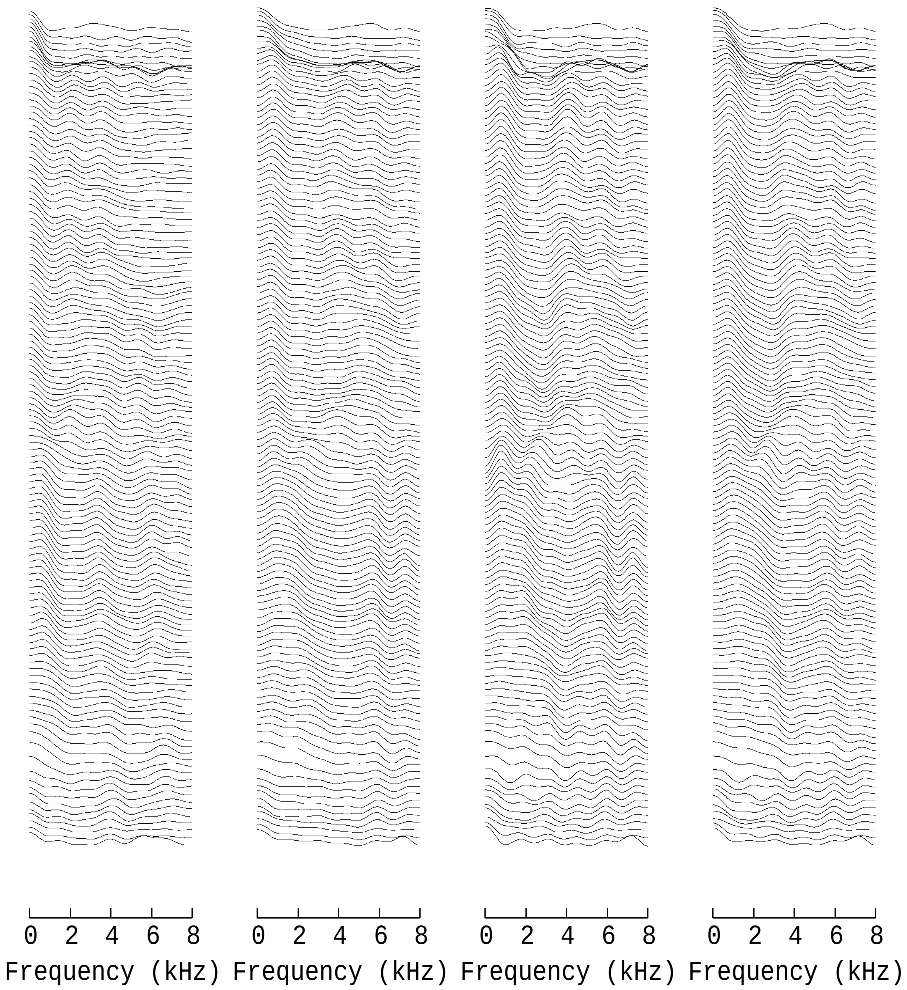
```
x2x +sf < data.short | frame -l 400 -p 80 | window -l 400 -L 512 |\
mgcep -m 12 -a 0.42 -c 2 -l 512 |\
mgc2mgc -m 12 -a 0.42 -c 2 -M 12 -A 0.42 -G 0 > data.mgcep.mcep
```

7.5 Plotting spectra for each speech recognition parameter

Files: data.lpc.cep: LPC cepstrum (float)
data.lpc.mcep: LPC mel-cepstrum (float)
data.mcep.mcep: mel-cepstrum (float)
data.mgcep.mcep: mel-cepstrum derived from mel-generalized cepstrum (float)

Conditions: plotted frames: from 10-th to 135-th

```
(\
bcut +f -n 12 -s 10 -e 135 < data.lpc.cep |\
mgc2sp -m 12 -a 0 -g 0 -l 512 |\
grlogsp -l 512 -x 8 -O 1 -c "(a) LPC-CEP" ;\
\
bcut +f -n 12 -s 10 -e 135 < data.lpc.mcep |\
mgc2sp -m 12 -a 0.42 -g 0 -l 512 |\
grlogsp -l 512 -x 8 -O 2 -c "(b) LPC-MCEP" ;\
\
bcut +f -n 12 -s 10 -e 135 < data.mcep.mcep |\
mgc2sp -m 12 -a 0.42 -g 0 -l 512 |\
grlogsp -l 512 -x 8 -O 3 -c "(c) MCEP" ;\
\
bcut +f -n 12 -s 10 -e 135 < data.mgcep.mcep |\
mgc2sp -m 12 -a 0.42 -g 0 -l 512 |\
grlogsp -l 512 -x 8 -O 4 -c "(d) MGCEP-MCEP" ) | xgr
```



(a) LPC-CEP (b) LPC-MCEP (c) MCEP (d) MGCEP-MCEP

8 Playing with the Vocoder Based on Mel-Cepstrum

8.1 High- or low-pitched voice

Files: data.mcep.high.syn: synthesized speech (float)
data.mcep.low.syn: synthesized speech (float)

```
sopr -m 0.4 data.pitch |\
excite -p 80 | mlsadf -m 20 -a 0.42 -p 80 data.mcep |\
tee data.mcep.high.syn | da +f -s 16
```

```
sopr -m 2 data.pitch |\
excite -p 80 | mlsadf -m 20 -a 0.42 -p 80 data.mcep |\
tee data.mcep.low.syn | da +f -s 16
```

8.2 Fast- or slow-speaking voice

Files: data.mcep.fast.syn: synthesized speech (float)
data.mcep.slow.syn: synthesized speech (float)

```
sopr -m 1 data.pitch |\
excite -p 40 | mlsadf -m 20 -a 0.42 -p 40 data.mcep |\
tee data.mcep.fast.syn | da +f -s 16
```

```
sopr -m 1 data.pitch |\
excite -p 160 | mlsadf -m 20 -a 0.42 -p 160 data.mcep |\
tee data.mcep.slow.syn | da +f -s 16
```

8.3 Hoarse voice

Files: data.mcep.hoarse.syn: synthesized speech (float)

```
sopr -m 0 data.pitch |\
excite -p 80 | mlsadf -m 20 -a 0.42 -p 80 data.mcep |\
tee data.mcep.hoarse.syn | da +f -s 16
```

8.4 Robotic voice

Files: data.mcep.robot.syn: synthesized speech (float)

```
train -p 200 -l -1 | mlsadf -m 20 -a 0.42 -p 80 data.mcep |\
tee data.mcep.robot.syn | da +f -s 16
```

8.5 Child-like or deep voice

Files: data.mcep.child.syn: synthesized speech (float)
data.mcep.deep.syn: synthesized speech (float)

```
sopr -m 0.4 data.pitch |\
excite -p 80 | mlsadf -m 20 -a 0.1 -p 80 data.mcep |\
tee data.mcep.child.syn | da +f -s 16
```

```
sopr -m 2 data.pitch |\
excite -p 80 | mlsadf -m 20 -a 0.6 -p 80 data.mcep |\
tee data.mcep.deep.syn | da +f -s 16
```

8.6 Various voices

Files: data.float: original speech (float)
data.mcep.syn: synthesized speech (float)
data.mcep.{ high, low, fast, slow, hoarse, robot, child, deep }.syn: synthesized speech (float)

```
da +f -v -s 16 data.float data.mcep.syn \
data.mcep.{high,low,fast,slow,hoarse,robot,child,deep}.syn
```

9 Speech Synthesis Based on HMM

9.1 Speech parameter generation from a sequence of HMMs

Files: sample.pdf: sequence of mean and variance corresponding to a state sequence included in this example (float, little endian)[3]
sample.mcep: mel-cepstrum generated from a sequence of HMMs (float)

Conditions: analysis order: 24
weight coefficients for calculating delta: $w(-1) = -0.5, w(0) = 0, w(1) = 0.5$
weight coefficients for calculating delta-delta: $w(-1) = 0.25, w(0) = -0.5, w(1) = 0.25$

Note: The state sequence is determined according to the state duration densities of the HMMs. The algorithm is not included in SPTK.

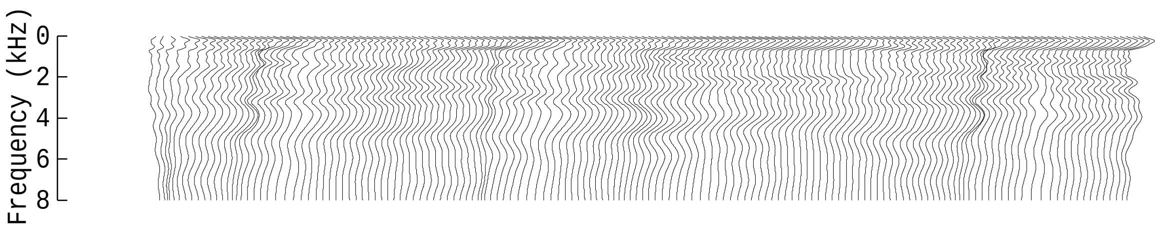
```
mlpg -m 24 -i 1 -d -0.5 0 0.5 -d 0.25 -0.5 0.25 sample.pdf > sample.mcep
```

9.2 Plotting spectra calculated from generated mel-cepstrum

Files: sample.mcep: mel-cepstral coefficients (float)

Conditions: analysis order: 24
frequency warping parameter: $\alpha = 0.42$
FFT size: 512 points
plotted frames: from 100-th to 250-th
sampling frequency: 16 kHz

```
bcut +f -n 24 -s 100 -e 250 < sample.mcep |\
mgc2sp -m 24 -a 0.42 -g 0 -l 512 | grlogsp -l 512 -x 8 -t | xgr
```



9.3 Speech synthesis from the generated mel-cepstrum

Files: sample.pitch: pitch data generated from a sequence of MSD-HMMs included in this example (float, little endian)[4]
sample.mcep: mel-cepstrum (float)
sample.mcep.syn: synthesized speech (float)

Conditions: frame period: 80 points (5 ms)
analysis order: 24
frequency warping parameter: $\alpha = 0.42$

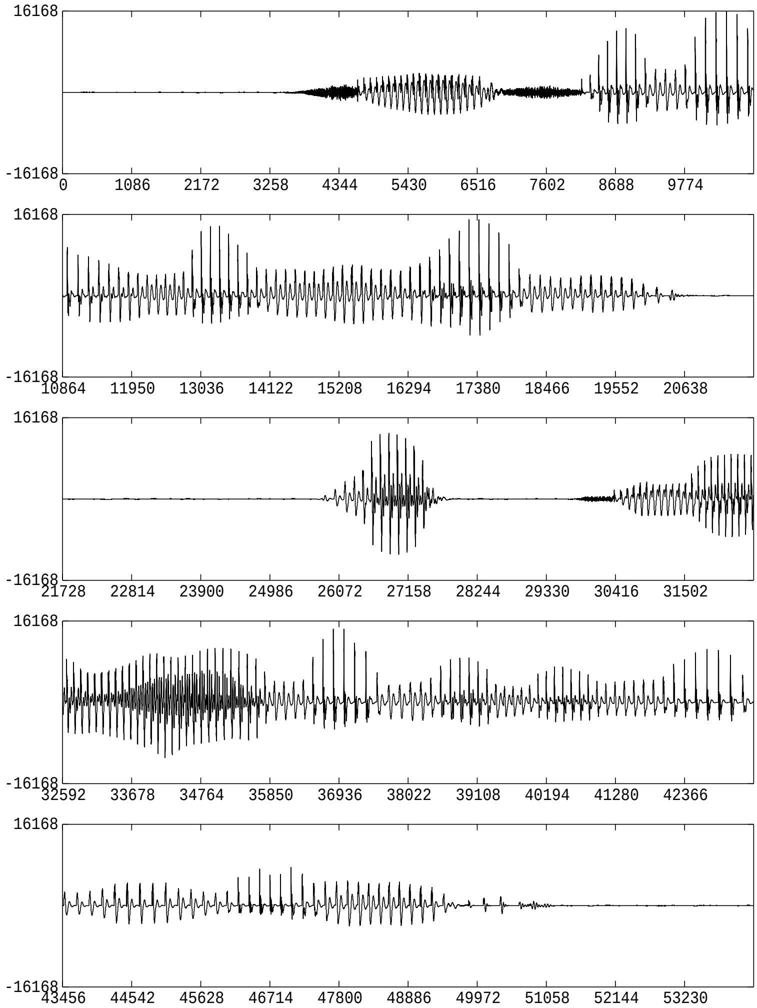
[3] If you compiled SPTK with "`--enable-double`" option, please first convert this file into double format:
`x2x +sd sample.pdf > sample.pdf.double`

[4] If you compiled SPTK with "`--enable-double`" option, please first convert this file into double format:
`x2x +sd sample.pitch > sample.pitch.double`

Note: The pitch pattern generation algorithm is not included in SPTK.

```
excite -p 80 sample.pitch |\
mlsadf -p 80 -a 0.42 -m 24 sample.mcep > sample.mcep.syn
```

```
gwave +f sample.mcep.syn | xgr
```




```
da +f -s 16 sample.mcep.syn
```

9.4 Check the given mean and variance vectors

Files: sample.pdf: sequence of mean and variance corresponding to a state sequence (float)

Conditions: analysis order: 24

9.4.1 Dump static feature vectors

```
bcp +f -l 150 -s 0 -e 24 sample.pdf | dmp -n 24 | less
```

9.4.2 Dump variance vectors of static feature vectors

```
bcp +f -l 150 -s 75 -e 99 sample.pdf | sopr -INV | dmp -n 24 | less
```

9.4.3 Dump dynamic feature vectors (delta)

```
bcp +f -l 150 -s 25 -e 49 sample.pdf | dmp -n 24 | less
```

9.4.4 Dump variance vectors of dynamic feature vectors (delta)

```
bcp +f -l 150 -s 100 -e 124 sample.pdf | sopr -INV | dmp -n 24 | less
```

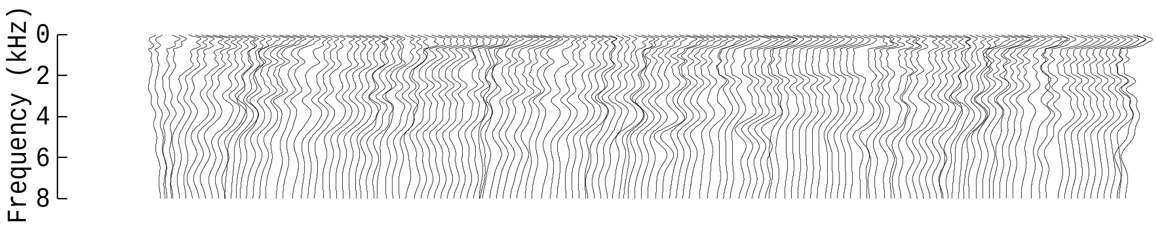
9.5 Speech synthesis without dynamic feature

Files: sample.pitch: pitch data generated from a sequence of MSD-HMMs (float)
sample.mcep.wo-dyn: mel-cepstrum generated without dynamic feature (float)
sample.mcep.wo-dyn.syn: synthesized speech without dynamic feature (float)

Conditions: frame period: 80 points (5 ms)
analysis order: 24
frequency warping parameter: $\alpha = 0.42$

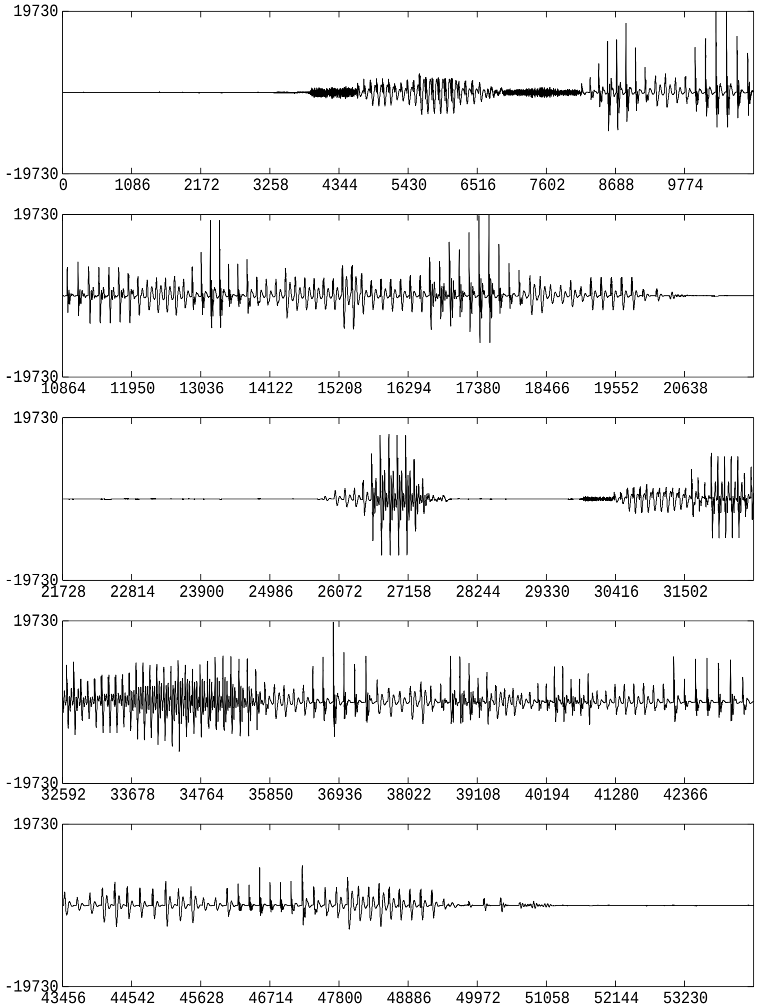
```
bcp +f -l 150 -s 0 -e 24 sample.pdf > sample.mcep.wo-dyn
```

```
bcut +f -n 24 -s 100 -e 250 < sample.mcep.wo-dyn |\
mgc2sp -m 24 -a 0.42 -g 0 -l 512 | grlogsp -l 512 -x 8 -t | xgr
```




```
excite -p 80 sample.pitch |\
mlsadf -p 80 -a 0.42 -m 24 sample.mcep.wo-dyn > sample.mcep.wo-dyn.syn
```

```
gwave +f sample.mcep.wo-dyn.syn | xgr
```




```
da +f -s 16 sample.mcep.wo-dyn.syn sample.mcep.syn
```

10 Voice Conversion based on GMM

Voice conversion from speaker maleA to speaker maleB

10.1 Minimum configuration of voice conversion

Files: source_maleA.short: original speech signal spoken by maleA (short integer, 16 kHz sampling, little endian)
target_maleB.short: target speech signal spoken by maleB (short integer, 16 kHz sampling, little endian)
test_maleA.short: test speech signal spoken by maleA (short integer, 16 kHz sampling, little endian)
converted_maleB.syn: converted speech signal (float)

Conditions: frame length: 400 points(25ms)
frame period: 80 points(5ms)
window: Blackman window
analysis order: 24

frequency warping parameter: α=0.42
the number of GMM mixture: 2

10.1.1 Training GMM

```
x2x +sf < source_maleA.raw | frame -l 400 -p 80 | window -l 400 -L 1024 |\
    mcep -l 1024 -m 24 -a 0.42 > source_maleA.mcep
x2x +sf < target_maleB.raw | frame -l 400 -p 80 | window -l 400 -L 1024 |\
    mcep -l 1024 -m 24 -a 0.42 > target_maleB.mcep
dtw -m 24 target_maleB.mcep < source_maleA.mcep | gmm -l 50 -m 2 -f > maleA_maleB.gmm
```

10.1.2 Voice conversion

```
x2x +sf < test_maleA.raw | frame -l 400 -p 80 | window -l 400 -L 1024 |\
    mcep -l 1024 -m 24 -a 0.42 > test_maleA.mcep
x2x +sf < test_maleA.raw | pitch -s 16 -p 80 > test_maleA.pitch
vc -n 24 -m 2 maleA_maleB.gmm < test_maleA.mcep > converted_maleB.mcep
excite -p 80 test_maleA.pitch |\
    mlsadf -m 24 -p 80 -a 0.42 converted_maleB.mcep > converted_maleB.syn
```

10.2 Voice conversion using iterative alignment

Files: source_maleA.short: original speech signal spoken by maleA (short integer, 16 kHz sampling, little endian)
target_maleB.short: target speech signal spoken by maleB (short integer, 16 kHz sampling, little endian)
test_maleA.short: test speech signal spoken by maleA (short integer, 16 kHz sampling, little endian)
converted_maleB_1.syn: converted speech signal (float)

Conditions: frame length: 400 points(25ms)
frame period: 80 points(5ms)
window : Blackman window
analysis order: 24
sampling frequency: 16kHz
frequency warping parameter: α=0.42
the number of GMM mixture: 2

10.2.1 Training initial GMM

```
dtw -m 24 target_maleB.mcep < source_maleA.mcep > maleA_maleB_0.dtw
gmm -l 50 -m 2 -f < maleA_maleB_0.dtw > maleA_maleB_0.gmm
```

10.2.2 GMM estimation using iterative alignment

```
x2x +sf < source_maleA.raw | frame -l 400 -p 80 | window -l 400 -L 1024 |\
    mcep -l 1024 -m 24 -a 0.42 |\
    vc -n 24 -m 2 maleA_maleB_0.gmm |\
    dtw -m 24 target_maleB.mcep -v maleA_maleB.viterbi > /dev/null
dtw -m 24 -V maleA_maleB.viterbi target_maleB.mcep < source_maleA.mcep > maleA_maleB_1.dtw
gmm -l 50 -m 2 -f < maleA_maleB_1.dtw > maleA_maleB_1.gmm
```

10.2.3 Voice conversion

```
vc -n 24 -m 2 maleA_maleB_1.gmm < test_maleA.mcep > converted_maleB_1.mcep
excite -p 80 test_maleA.pitch |\
    mlsadf -m 24 -p 80 -a 0.42 converted_maleB_1.mcep > converted_maleB_1.syn
```

11 Speaker Identification Based on GMM

identification of speaker maleB from speaker maleA, maleB and maleC

Files: data_male{A,B,C}.short: speech signal spoken by maleA,B and C (short integer, 16 kHz sampling, little endian)
test_maleB.short: test speech signal spoken by maleB (short integer, 16 kHz sampling, little endian)

Conditions: order of mfcc: 12

11.1 GMM training

```
x2x +sf < data_maleA.short | frame | mfcc | gmm -l 12 > maleA.gmm
x2x +sf < data_maleB.short | frame | mfcc | gmm -l 12 > maleB.gmm
x2x +sf < data_maleC.short | frame | mfcc | gmm -l 12 > maleC.gmm
```

11.2 Speaker identification

```
x2x +sf < test_maleB.short | frame | mfcc > test_maleB.mfcc
gmmp -a -l 12 maleA.gmm test_maleB.mfcc > result_maleA.score
gmmp -a -l 12 maleB.gmm test_maleB.mfcc > result_maleB.score
gmmp -a -l 12 maleC.gmm test_maleB.mfcc > result_maleC.score
```

The recognized speaker's score is the largest value for the test speech signal.